REMARKS

Claims 10-18 are pending in the application.

Allowable Subject Matter

Applicants appreciate the Examiner's indication that claims 17 and 18 are allowable over the prior art of record.

An Embodiment of the Present Invention

An embodiment of the present invention is directed to an apparatus for encoding an audio signal to obtain an encoded audio signal to be used by a decoder having a high-frequency reconstruction module for performing a high-frequency reconstruction for a frequency range above a crossover frequency. As shown in Fig. 5, the apparatus includes, a core encoder 502 for encoding a lower frequency band of the audio signal up to the crossover frequency. The core encoder has a variable crossover frequency which is controllable with respect to the variable crossover frequency, and operable on a block-wise frame by frame basis. The apparatus also includes a crossover frequency control module 504 for estimating, dependent on at least one of a measure of the degree of difficulty for encoding the audio signal by the core encoder 502 and a border between a tonal and a noise-like frequency range of the audio signal, the crossover frequency to be selected by the core encoder for a frame of a series of subsequent frames, so that the crossover frequency is variable adaptively over time for the series of subsequent frames. The crossover frequency control module 504 is adapted to control the core encoder with respect to the crossover frequency.

Claim Rejections – 35 U.S.C. § 103

(a) Claims 10, 11, and 16 have been rejected under 35 U.S.C. § 103(a) as being unpatentable over Taniguchi et al. ("A High-Efficiency Speech Coding Algorithm based on ADPCM with Multi-Quantizer," International Conference on Acoustics, Speech, and Signal Processing, April 1986) in view of Kamai et al. (USP 6,490,562). This rejection is respectfully traversed.

Taniguchi discloses an encoding algorithm in the left half of Fig. 1. An enlarged copy of Figs. 1 and 2 are attached herewith. As stated in the first two lines below Fig. 1, an input signal is divided into three sub-band, namely lower, middle, and upper bands using two-stage Quadrature Mirror Filters (QMFs). The lower band output by QMF-1 is fed to a ADPCM-MQ coder. Thus, bandwidth and, particularly the upper frequency limit of this lower band is fixed and determined by the QMF-1. When the upper limit of the lower band in Fig. 1, i.e., the highest frequency, which is included in the lower band is compared to the "crossover frequency" as defined in amended claim 10, the ADPCM-MQ coder in Fig. 1, which is shown in detailed in Fig. 2 can be regarded as the "core encoder." However, the core encoder in Taniguchi does not have a variable crossover frequency.

Instead, the crossover frequency is determined by the QMF-1.

Further, the core encoder in document Taniguchi is controllable with respect to the variable crossover frequency. However, in Taniguchi, the QMF-1, which determines the upper limit of the lower band is not controllable but is fixedly set.

Finally, Taniguchi also does <u>not</u> disclose a crossover frequency control module since there is no module in Fig. 1, which would calculate any control signal for the QMF-1, since the QMF-1 is not controllable and is not controlled to change the upper frequency limit of the lower band as shown in Fig. 1 and Fig. 2, or to –generally- change the crossover frequency.

Next, Applicants respectfully disagree with Examiner's interpretation of Taniguchi regarding equation 3.

As can be seen in Fig. 2, the ADPCM-MQ codec includes m different ADPCM codecs (i.e., "ADPCM-1," "ADPCM-2," . . . "ADPCM-m"). Each of these different encoders has a different quantizer characteristics as it is outlined in the forth line below equation (Eq.2).

However, in each case, the residual signal output by such an ADPCM device covers exactly the same bandwidth of the original lowband. Stated in other words, the ADCPM codecs are different with respect to their quantization step size Δ m(n) as outlined in the second line below equation (Eq.2).

Thus, each and every residual signal output by an ADPCM device covers exactly the same bandwidth of the original lowband. Therefore, irrespective of the selection preformed by equation 3 on page 1722 in Taniguchi, the selected (the selection is performed by the selector in Fig. 2) residual signal multiplexed into the bit stream by the multiplexer (MUC) covers always the same bandwidth or has the same crossover frequency, which is determined by the <u>fixed</u> QUM-1 device as shown in Fig. 1.

Therefore, Taniguchi discloses a quantizer step size control module rather than a <u>crossover frequency</u> control module.

Further, as shown equation 3 and line 9 of the paragraph above equation 3, the selection of a certain ADPCM coder among the m ADPCM codecs is performed based on the "power difference between the locally decoded signal in each ADPCM coding block and the input signal (the quantization error power).

However, a quantization error has no relationship whatsoever with a degree of difficulty for encoding the audio signal with a border between a tonal and a noise-like frequency range of the audio signal.

In other words, a quantization error introduced by a quantizer having a certain quantizer step size is completely uncorrelated to the degree of difficulty for encoding or a border between a tonal and a noise-like frequency range as recited in claim 10.

Thus, the Examiner's evaluation of Taniguchi in the first paragraph of page 2 of the Office Action is incorrect.

With regard to Kamai, as stated in the Reply filed on December 9, 2004, the cut-off frequency or center frequency of the adaptive filter in Kamai is only used to extract some pitch mark information. However, as recited in claim 10, this value derived in Kamai has no relationship whatsoever with the crossover frequency, which defines the lower frequency band encoded by the core encoder.

Therefore, Applicants respectfully submit that the Examiner is not correct in stating that the cut-off or center frequency is the same as the crossover frequency as recited in claim 10.

Additionally, Taniguchi shows, in Fig. 2, an audio coder, which performs some quantization, wherein the kind of quantization, i.e., the optimum ADPCM coding block is

determined by equation 3 of Taniguchi, while Kamai is a fully parametric speech encoder, which does not use any quantization, but fully parameterizes an input signal and send some parameters to a decoder or speech synthesizer, which performs a kind of table look-up using the transmitted information in order to overlay pre-stored wave forms. Therefore, the quantization-based encoder in Taniguchi and the parametric speech processor in Kamai are in contrast to each other and exclude each other. In other word, one teaches away the other. Therefore, it would not make sense to use any information from Kamai with respect to the pitch mark information for quantization purpose as disclosed in Taniguchi.

Therefore, even assuming, *arguendo*, that Taniguchi and Kamai can be combined, Taniguchi in view of Kamai fails to disclose or even suggest the "crossover frequency control module" as recited in claim 10.

Claim 16 is allowable at least for the similar reasons as stated in the foregoing with respect to claim 10.

The Examiner is respectfully requested to reconsider and withdraw this rejection.

(b) Claims 14 and 15 have been rejected under 35 U.S.C. § 103(a) as being unpatentable over Taniguchi in view of Kamai, and further in view of Shoham et al. (USP 5,646,961). This rejection is respectfully traversed.

Claims 14 and 15, dependent on claim 10, are allowable at least for their dependency on claim 10.

The Examiner is respectfully requested to reconsider and withdraw this rejection.

Conclusion

Accordingly, in view of the above amendments and remarks, reconsideration of the rejections and objections, and allowance of the pending claims are earnestly solicited.

Should there be any outstanding matters that need to be resolved in the present application, the Examiner is respectfully requested to contact Maki Hatsumi (#40,417) at the telephone number of the undersigned below, to conduct an interview in an effort to expedite prosecution in connection with the present application.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or to credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under 37 C.F.R. § 1.17; particularly, extension of time fees.

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Respectfully submitted,

By

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Attachments: Enlarged/Commented Figures 1 & 2 of Taniguchi et al. reference

BASED ON ADPCM WITH MULTI-QUANTIZER

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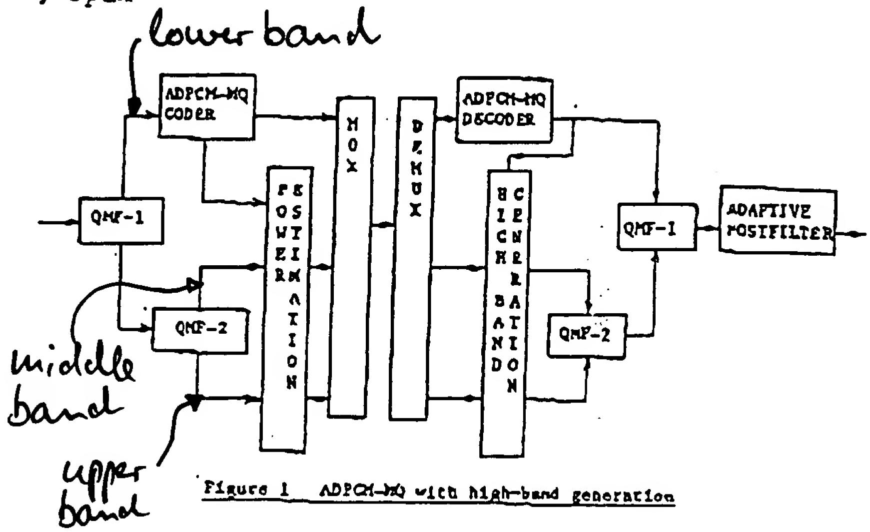

Figure 1 ADPCM-MQ with high-band generation

The input signal is divided into 3 sub-bands, namely the lower, middle and upper band, by two-stage Quadrature Mirror Filters (QMFs). The ratio of the sub-band widths is 2:1:1. The ADPCM-MQ coding is carried out on the lower band signal only. For the middle and upper bands, the signal power in the frame is calculated, and the amplitude ratio of the signal in each band is determined according to the signal power of the locally decoded lower-band signal. The amplitude ratio is transmitted as additional information, so that the middle and upper band signals can be reconstructed from the lower band signal at the decoder. The reconstructed speech signal is enhanced by an adaptive postfilter[5] which uses the coefficients of the lower band ADPCM predictor. Computer simulations and subjective tests of this algorithm have indicated good reproduced speech quality.

ADPCM-MQ ALGORITHM

Because the number of residual signal quantizing bits is two or less, quantizing noise prevents ADPCM from giving good speech quality. As one solution to this problem, the authors have proposed an ADPCM-MQ coding algorithm in which several ADPCM codecs with different quantizing step-size update rates are driven in parallel, the quantization error power is evaluated in each frame, and the quantizer which provides the best S/N ratio is found and selected dynamically.(Figure 2) Since the ADPCM adaptive predictor and adaptive quantizer are located in a same feedback loop subject to constant adaptive control, the prediction characteristic of the adaptive predictor cannot be separated from the quantizing characteristic of the adaptive quantizer. Improvement of the quantizing characteristic by switching the adaptive quantizer in response to the characteristics of the input signal can also be

lowerband (upper frequency limit is fixed and determined by QMF-1)

different quantizers selected by equation 3

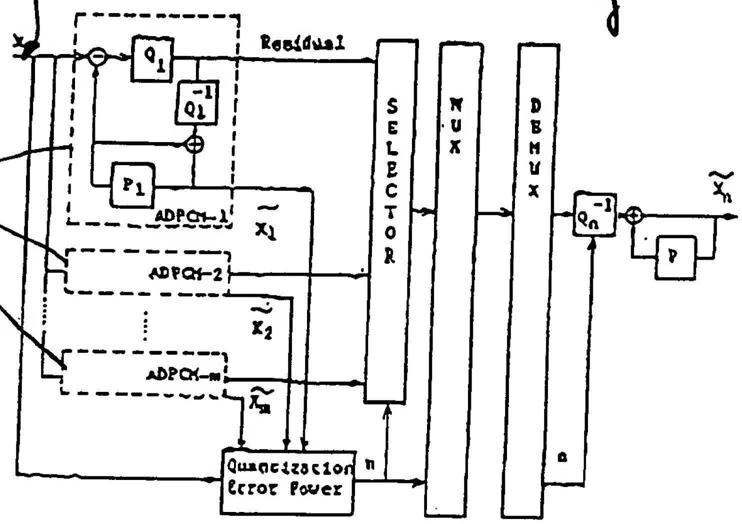

Figure 2 Structure of ADPCM-MQ CODEC

In this algorithm, instead of the single step-size update formula (Equation 1) of the conventional ADPCM quantizer[6], several quantizers with different coefficient values $\mu m(Im(n))$ indicated in Equation 2 are used.

$$\Delta(n+1) = \Delta(n)^{r} * M(I(n))$$

$$\log2 \Delta(n+1) = r*\log2 \Delta(n) + \log2 M(I(n))$$

$$D(n+1) = r*D(n) + \mu(I(n)) \qquad \text{(Eq.1)}$$

$$Dm(n+1) = r*Dm(n) + \mu m(Im(n)) \qquad \text{(Eq.2)}$$

In this equation $Dm(n)$ is the logarithm of the quantization step-size $\Delta m(n)$ of the m-th quantizer at time n, and $Im(n)$ is the quantizer output code. Since their adaptive quantizers have different characteristics, the adaptive predictors of each ADPCM coding block operate differently under the feedback control. To select the optimum ADPCM encoder, we use the function given by Equation 3, which stands for the power difference between the locally decoded signal in each ADPCM coding block and the input signal (the quantization error power). Thus, the optimum ADPCM coding block is selected for each frame. To prevent discontinuities at the frame boundary, the coefficients and tap data of adaptive predictor and quantization step-size corresponding to the selected adaptive quantizer are copied to the other predictors and quantizers before the next frame signals are processed. Figures 3 and 4 show the simulation results of the improvement in the ADPCM codec characteristics obtained by using the multi-quantizer (MQ) scheme. Figure 3 shows the relationship between frame length and segmental S/N ratio. Figure 4 shows the relationship between the number of quantizers and the segmental S/N ratio.

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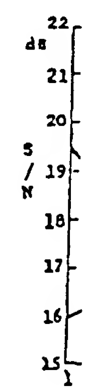

Figure 3 Seg